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INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04B 1/66, H04L 5/06	A1	(11) International Publication Number: WO 98/09383 (43) International Publication Date: 5 March 1998 (05.03.98)

(21) International Application Number: PCT/CA97/00608

(22) International Filing Date: 29 August 1997 (29.08.97)

(30) Priority Data:
2,184,541 30 August 1996 (30.08.96) CA

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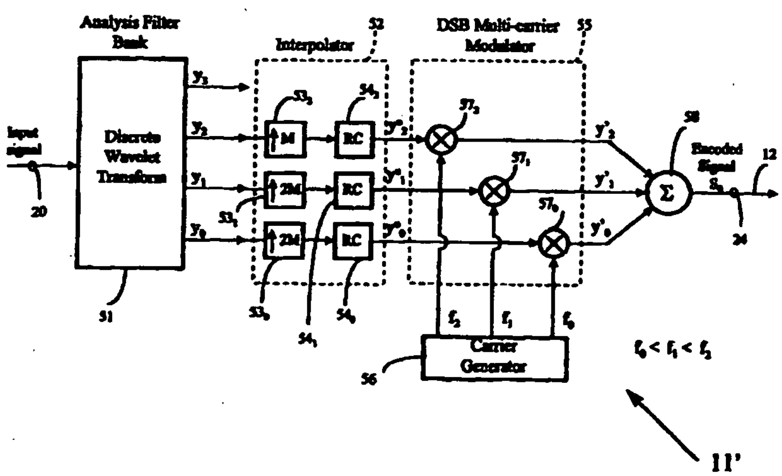
(81) Designated States: AT, AU, BR, CA, CN, CZ, DE, ES, FI, GB, HU, JP, KR, MX, NZ, RU, SE, SG, UA, US, European patent (AT, BE, CH, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: FREQUENCY DIVISION MULTIPLEXED TRANSMISSION OF SUB-BAND SIGNALS



(57) Abstract

A method and apparatus for processing an input signal for transmission and/or storage, uses an analysis filter bank (21; 51) to decompose the signal into sub-band signals which are used to modulate a plurality of carriers. The carriers are combined into a single encoded signal for transmission/storage. The encoder/decoder is especially applicable to telecommunications systems and recording systems. The analysis filter bank may comprise a multiresolution filter, such as an octave band filter bank (40A/B/C/D...43A/B/C/D) implementing Discrete Wavelet Transform. The modulation may comprise double-sideband, single-sideband, quadrature amplitude modulation, and so on. Where the input signal is analog, the carriers may be modulated directly by the sub-band signals. Where the input signal is digital, however, the sub-band signals are interpolated, all to the same rate, and then used to modulate the carriers. The corresponding decoder (13) extracts the modulated carrier signals, demodulates them, decimates them (if applicable) and then synthesizes them to reconstruct the original input signal. One or more of the sub-bands, especially at the higher frequencies, may be discarded. Discrete wavelet transformation is applied to segments of the digital signal.

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FREQUENCY DIVISION MULTIPLEXED TRANSMISSION OF SUB-BAND SIGNALS

BACKGROUND OF THE INVENTION

TECHNICAL FIELD

The invention relates to a method and apparatus for encoding signals, whether digital or analog, for transmission and/or storage. The invention is especially, but not exclusively, applicable to the encoding of digital signals for transmission via communications channels, such as twisted wire pair subscriber loops in telecommunications systems or to storage of signals in or on a storage medium, such as video signal recordings, audio recordings, data storage in computer systems, and so on.

BACKGROUND ART

Embodiments of the invention are especially applicable to Asynchronous Transfer Mode (ATM) telecommunications systems. Such systems are now available to transmit millions of data bits in a single second and are expected to turn futuristic interactive concepts into exciting realities within the next few years. However, deployment of ATM is hindered by expensive port cost and the cost of running an optical fiber from an ATM switch to the customer-premises using an architecture known as Fiber-to-the-home. Running ATM traffic in part of the subscriber loop over existing copper wires would reduce the cost considerably and render the connection of ATM to customer-premises feasible.

The introduction of ATM signals in the existing twisted-pair subscriber loops leads to a requirement for bit rates which are higher than can be achieved with conventional systems in which there is a tendency, when transmitting at high bit rates, to lose a portion of the signal, typically the higher frequency part, causing the signal quality to suffer significantly. This is particularly acute in two-wire subscriber loops, such as so-called twisted wire pairs. Using quadrature amplitude modulation (QAM), it is possible to meet the requirements for Asymmetric Digital Subscriber Loops (ADSL), involving rates as high as 1.5 megabits per second for loops up to 3 kilometers long with specified error rates. It is envisaged that ADSL systems will allow rates up to about 8 megabits per second over 1 kilometer loops. Nevertheless, these rates are still considered to be too low, given that standards currently proposed for ATM basic

subscriber access involve rates of about 26 megabits per second.

QAM systems tend to operate at the higher frequency bands of the channel, which is particularly undesirable for two-wire subscriber loops where attenuation and cross-talk are worse at the higher frequencies. It has been proposed, therefore, to use frequency division modulation (FDM) to divide the transmission system into a set of frequency-indexed sub-channels. The input data is partitioned into temporal blocks, each of which is independently modulated and transmitted in a respective one of the sub-channels. One such system, known as discrete multi-tone transmission (DMT), is disclosed in United States patent specification No. 5,479,447 issued December 1995 and in an article entitled "Performance Evaluation of a Fast Computation Algorithm for the DMT in High-Speed Subscriber Loop", IEEE Journal on Selected Areas in Communications, Vol. 13, No. 9, December 1995 by I. Lee *et al.* Specifically, US 5,479,447 discloses a method and apparatus for adaptive, variable bandwidth, high-speed data transmission of a multi-carrier signal over a digital subscriber loop. The data to be transmitted is divided into multiple data streams which are used to modulate multiple carriers. These modulated carriers are converted to a single high speed signal by means of IFFT (Inverse Fast Fourier Transform) before transmission. At the receiver, Fast Fourier Transform (FFT) is used to split the received signal into modulated carriers which are demodulated to obtain the original multiple data streams.

Such a DMT system is not entirely satisfactory for use in two-wire subscriber loops which are very susceptible to noise and other sources of degradation which could result in one or more sub-channels being lost. If only one sub-channel fails, perhaps because of transmission path noise, the total signal is corrupted and either lost or, if error detection is employed, may be retransmitted. It has been proposed to remedy this problem by adaptively eliminating noisy sub-channels, but to do so would involve very complex circuitry.

A further problem with DMT systems is poor separation between sub-channels. In United States patent specification No. 5,497,398 issued March 1996, M.A. Tzannes and M.C. Tzannes proposed ameliorating the problem of degradation due to sub-channel loss, and obtaining superior burst noise immunity, by replacing the Fast Fourier Transform with a lapped transform, thereby increasing the difference between the main lobe and side lobes of the filter response in each sub-channel. The lapped transform may comprise wavelets, as disclosed by M.A. Tzannes, M.C. Tzannes and H.L. Resnikoff

in an article "The DWMT: A Multicarrier Transceiver for ADSL using *M*-band Wavelets", ANSI Standard Committee T1E1.4 Contribution 93-067, Mar. 1993 and by S.D. Sandberg, M.A. Tzannes in an article "Overlapped Discrete Multitone Modulation for High Speed Copper Wire Communications", IEEE Journal on Selected Areas in Comm., Vol. 13, No. 9, pp. 1571-1585, Dec. 1995, such systems being referred to as "Discrete Wavelet Multitone (DWMT).

A disadvantage of both DMT and DWMT systems is that they typically use a large number of sub-channels, for example 256 or 512, which leads to complex, costly equipment and equalization and synchronization difficulties. These difficulties are exacerbated if, to take advantage of the better characteristics of the two-wire subscriber loop at lower frequencies, the number of bits transmitted at the lower frequencies is increased and the number of bits transmitted at the higher frequencies reduced correspondingly.

It is known to use sub-band filtering to process digital audio signals prior to recording on a storage medium, such as a compact disc. Thus, US patent specification number 5,214,678 (Rault *et al*) discloses an arrangement for encoding audio signals and the like into a set of sub-band signals using a commutator and a plurality of analysis filters, which could be combined. Rault *et al* use recording means which record the sub-band signals as multiple, distinct tracks. This is not entirely satisfactory because each sub-band signal would require its own recording head or, if applied to transmission, its own transmission channel.

United States patent specification number 5,161,210 (Druyvesteyn) discloses a similar analysis technique to that disclosed by Rault *et al* but, in this case, the sub-band signals are combined by means of a synthesis filter before recordal. The input audio signal first is analyzed, and an identification signal is mixed with each of the sub-band signals. The sub-band signals then are recombined using a synthesis filter. The technique ensures that the identification signal cannot be removed simply by normal filtering. The frequency spectrum of the recombined signal is substantially the same as that of the input signal, so it would still be susceptible to corruption by loss of the higher frequency components. The corresponding decoder also comprises an analysis filter and a synthesis filter. Consequently, the apparatus is very complex and would involve delays which would be detrimental in high speed transmission systems.

It is desirable to combine the sub-band signals in such a way as to reduce the risk

of corruption resulting from part of the signal being lost or corrupted during transmission and/or storage.

It should be noted that, although Rault *et al* use the term "analysis filter" in their specification, in this specification the term "analysis filter" will be used to denote a device which decomposes a signal into a plurality of sub-band signals in such a way that the original signal can be reconstructed using a complementary synthesis filter.

SUMMARY OF THE INVENTION:

The present invention seeks to eliminate, or at least mitigate, the disadvantages of these known systems and has for its object to provide an improved method and apparatus for encoding signals for transmission and/or storage.

According to one aspect of the invention, apparatus for encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, characterized by:

an encoder comprising

(i) analysis filter means for analyzing the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of different frequencies;

(ii) means for providing a plurality of carrier signals having different frequencies;

(iii) modulation means for using at least some of the sub-band signals each to modulate a respective one of the plurality of carrier signals; and

(iv) combining means for combining the modulated carrier signals to form the encoded signal for transmission or storage;

and a decoder comprising

(v) filter means for extracting the plurality of modulated carrier signals from the received or recorded encoded signal;

(vi) means for providing a plurality of carrier signals corresponding to those of said encoder;

(vii) demodulation means for using at least some of the carrier signals to demodulate each of the extracted modulated carrier signals to extract the corresponding plurality of sub-band signals; and

(vii) synthesis filter means complementary to said analysis filter means for

processing the plurality of extracted sub-band signals to produce a decoded signal corresponding to the input signal.

According to second and third aspects of the invention, there are provided the encoder *per se* and the decoder *per se* of the apparatus.

The analysis filter means may be uniform, for example an M-band filter bank, or non-uniform, for example a "multiresolution" filter bank such as an octave-band or dyadic filter bank which will produce sub-bands having different bandwidths, typically each half the width of its neighbour, or even a Short-time Fast Fourier Transform unit.

Where the input signal is digital, the analysis filter means may comprise an octave band filter bank implementing discrete wavelet transform (DWT).

For encoding digital input signals, the encoder may further comprise interpolation means for providing, for each sub-band signal, interpolated values between adjacent actual values of the sub-band signal. The decoder then will comprise a decimator for decimating the extracted sub-band signals to remove interpolated values before application to the synthesis filter means. The interpolation rate will be in proportion to the sampling rate of the sub-band concerned, so that the resulting interpolated sub-band signals all have the same rate.

The interpolation rate will be chosen according to the requirements of a particular transmission channel or storage means but typically will be of the order of 1:8 or more.

The interpolation means may comprise an upsampler, for interpolating intervals between actual values, and filter means, for example Raise Cosine filter means, for determining values between the actual samples and inserting them at the appropriate intervals.

Usually, when used with digital signals, sub-band analysis filters create sub-band signals which occupy a wide spectrum as compared with the original signal, which makes modulation difficult. Interpolating and smoothing the sub-band signals advantageously band-limits the spectrum of the sub-band signals, permitting modulation by a variety of techniques, for example Double or Single Sideband Amplitude Modulation, Quadrature Amplitude Modulation (QAM), Carrier Amplitude/Phase modulation (CAP), and so on.

According to a fourth aspect of the invention, there is provided a method of encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, <u>characterized in that</u> the encoding of the input signal comprises the steps of:

(i) using analysis filter means, analyzing the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of frequencies;

(ii) providing a plurality of carrier signals having different frequencies;

(iii) using at least some of the sub-band signals each to modulate a respective one of the plurality of carrier signals; and

(iv) combining the modulated carrier signals to form the encoded signal for transmission or storage;

and the decoding of the encoded signal comprises the steps of:

(v) extracting the plurality of modulated carrier signals from the received or recorded encoded signal;

(vi) providing a plurality of carrier signals corresponding to those used in said encoding;

(vii) using at least some of the carrier signals to demodulate the extracted modulated carrier signals to extract the corresponding plurality of sub-band signals; and

(vii) using synthesis filter means complementary to said analysis filter means, processing the plurality of extracted sub-band signals to produce a decoded signal corresponding to the input signal.

According to fifth and sixth aspects of the invention, there are provided the method of encoding *per se* and the method of decoding *per se*.

In embodiments of any of the above aspects of the invention which use Discrete Wavelet Transform, the digital input signal may be divided into segments and the discrete wavelet transform used to transform successive segments of the digital signal.

Where the input signal is analog, it may be digitized and sampled, then segmented (if appropriate). Where an octave band filter bank is used to implement DWT, however, segmentation of the input signal is not needed. Alternatively, an analog analysis filter bank could be used operating upon the continuous analog signal directly, each sub-baud signal comprising a different centre frequency.

The foregoing and other objects, features, aspects and advantages of the present invention will become more apparent from the following detailed description of preferred embodiments of the invention which are described by way of example only with

reference to with the accompanying drawings.

BRIEF DESCRIPTION OF DRAWINGS:

Figure 1 is a simplified schematic diagram illustrating a transmission system including an encoder and decoder according to the invention;

Figure 2 is a schematic block diagram of an encoder embodying the present invention;

Figure 3 is a schematic block diagram of a corresponding decoder for signals from the encoder of Figure 1;

Figure 4A illustrates three-stage Discrete Wavelet Transform decomposition using a pyramid algorithm to provide sub-band signals;

Figure 4B illustrates three-stage synthesis of an output signal from the sub-band signals of Figure 4A;

Figure 5 is a block schematic diagram of an encoder using a sub-band analysis filter and Double Sideband amplitude modulation with three sub-bands and corresponding carriers;

Figure 6 is a block schematic diagram of a decoder using three sub-bands and carriers for use with the encoder of Figure 5;

Figures 7A, 7B and 7C illustrate the frequency spectrums of an input signal, and three sub-band signals before and after multi-carrier SSB modulation;

Figure 8 illustrates, as an example, a very simple input signal S_i applied to the encoder of Figure 5;

Figures 9A, 9B, 9C and 9D illustrate the sub-band wavelet signals y_0, y_1, y_2 and y_3 produced by analysis filtering of the input signal S_i of Figure 8;

Figures 10A, 10B and 10C illustrate modulated carrier signals y'_0, y'_1 and y'_2 modulated by sub-band wavelet signals y_0, y_1 and y_2, respectively;

Figure 11 illustrates the encoded/transmitted signal S_o;

Figure 12 illustrates the power spectrum of the transmitted signal S_o;

Figures 13A, 13B and 13C illustrate the recovered wavelet modulated carriers, y''_0, y''_1 and y''_2

Figure 14A, 14B and 14C illustrate the recovered wavelet signals y^*_0, y^*_1 and y^*_2; and

Figure 15 illustrates the reconstructed signal.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

A transmission system embodying the present invention is illustrated in Figure 1. The system comprises digital input signal source **10**, an encoder **11**, transmission medium **12**, decoder **13** and signal destination **14**. Input signal S_i from signal source **10** is applied to the encoder **11** which encodes it using sub-band filtering and multi-carrier modulation and supplies the resulting encoded signal S_o to transmission medium **12**, which is represented by a transmission channel **15**, noise source **16** and summer **17**, the latter combining noise with the signal in the transmission channel **15** before it reaches the decoder **13**. Although a transmission medium **12** is illustrated, it could be an analogous storage medium instead. The output of the decoder **13** is supplied to the signal destination **14**. The usable bandwidth of channel **15** dictates the maximum allowable rate of a signal that could be transmitted over the channel.

A first embodiment of the encoder **11** is illustrated in more detail in Figure 2. The input signal S_i is applied via an input port **20** to analysis filter bank **21** which decomposes it into sub-bands to generate/extract a lowpass sub-band signal y_0, bandpass sub-band signals y_1 - y_{N-2} and a highpass sub-band signal y_{N-1}. The sub-band signals y_1 - y_{N-1} are supplied to a multi-carrier modulator **22** which uses each sub-band signal to modulate a respective carrier of a selected frequency, as will be explained later. The lowpass sub-band signal y_0 contains more low frequency components than the other sub-band signal, and is used to modulate a low frequency carrier f_0; The bandpass sub-band signals y_1 - y_{N-2} and highpass sub-band signal y_{N-1} have more high frequency components than the lowpass wavelet signal y_0 and are therefore used to modulate higher frequency carrier signals f_1 - f_{N-1}, respectively, of which the frequencies increase from f_1 to f_{N-1}. The modulated carrier signals y'_0 - y'_{N-1} are combined by summer **23** to form the encoded output signal S_o which is transmitted via output port **24** to transmission medium **12** for transmission to decoder **13** (Figure 1).

A suitable decoder **13**, for decoding the encoded output signal, will now be described with reference to Figure 3. After passing through the transmission medium **12**, the transmitted signal S_o may be attenuated and contain noise. Hence, as received by the decoder at port **30** it is identified as received signal S'_o (the prime signifying that it is not identical to enclosed signal S_o) and supplied to a filter array **31**. Each of the filters in the array **31** corresponds to one of the frequencies f_o - f_{N-1} of the multi-carrier modulator 22 (Figure 2) and recovers the corresponding modulated carrier signals. The

recovered modulated carrier signals y''_0 - y''_{N-1} separated by the array then are demodulated by a multi-carrier demodulator **32** to recover the lowpass, bandpass and highpass sub-band signals y^*_0 - y^*_{N-1} corresponding to sub-band signals y_0 - $y_{(N-1)}$, respectively, in the encoder **11**. The recovered sub-band signals are supplied to synthesis filter bank **33** which, operating in a complementary and inverse manner to analysis filter bank **21**, produces an output signal S'_i which should closely resemble the input signal S_i in Figure 2, and supplies it to signal destination **14** via output port **34**. Usually, the recovered signal S'_i will be equalized using an adaptive equalizer to compensate for distortion and noise introduced by the channel **12**.

It should be noted that the highpass sub-band signal y_{N-1} and some of sub-band signals y_0 - y_{N-2} in Figure 2 may not need to be transmitted, if they contain little transmission power as compared with other sub-band signals. When these sub-band signals are not transmitted, the synthesis filter bank **33** shown in Figure 3 will insert zeros in place of the missing sub-band signals. The reconstructed signal S'_i would then be only a close approximation to the original input signal S_i. Generally, the more sub-bands used, the better the approximation.

Preferably, analysis filter **21** is a multiresolution filter bank which implements a Discrete Wavelet Transform (DWT). In order to facilitate a better understanding of the embodiments which use DWT, a brief introduction to discrete wavelet transforms (DWT) will first be given. DWT represents an arbitrary square integrable function as the superposition of a family of basis functions called *wavelets*. A family of wavelet basis functions can be generated by translating and dilating the *mother wavelet* corresponding to the family. The DWT coefficients can be obtained by taking the inner product between the input signal and the wavelet functions. Since the basis functions are translated and dilated versions of each other, a simpler algorithm, known as *Mallat's tree algorithm* or *pyramid algorithm*, has been proposed by S. G. Mallat in "A theory of multiresolution signal decomposition: the wavelet representation", *IEEE Trans. on Pattern Recognition and Machine Intelligence*, Vol. 11, No. 7, July 1989. In this algorithm, the DWT coefficients of one stage can be calculated from the DWT coefficients of the previous stage, which is expressed as follows:

$$W_L(n,j) = \sum_m W_L(m,j-1)\,h(m-2n) \tag{1a}$$

$$W_H(n,j) = \sum_m W_L(m,j-1)\, g(m-2n) \qquad \text{(1b)}$$

where $W(p,q)$ is the p-th wavelet coefficient at the q-th stage, and $h(n)$ and $g(n)$ are the dilation coefficients corresponding to the scaling and wavelet functions, respectively.

For computing the DWT coefficients of the discrete-time data, it is assumed that the input data represents the DWT coefficients of a high resolution stage. Equations 1a and 1b can then be used for obtaining DWT coefficients of subsequent stages. In practice, this decomposition is performed only for a few stages. It should be noted that the dilation coefficients $h(n)$ represent a lowpass filter, whereas the coefficients g(n) represent a highpass filter. Hence, DWT extracts information from the signal at different scales. The first stage of wavelet decomposition extracts the details of the signal (high frequency components) while the second and all subsequent stages of wavelet decompositions extract progressively coarser information (lower frequency components). It should be noted that compactly supported wavelets can be generated by a perfect-reconstruction two-channel filter banks with a so-called octave-band tree-structured architecture. Orthogonal and biorthogonal filter banks can be used to generate wavelets in these system. A three stage octave-band tree structure for Discrete Wavelet Transformation will now be described with reference to Figures 4A and 4B, in which the same components in the different stages have the same reference number but with the suffix letter of the stage.

Referring to Figure 4A, the three decomposition stages A, B and C have different sampling rates. Each of the three stages A, B and C comprises a highpass filter **40** in series with a downsampler **41**, and a lowpass filter **42** in series with a downsampler **43**. The cut-off frequency of each lowpass filter **42** is substantially the same as the cut-off frequency of the associated highpass filter **40**. In each stage, the cut-off frequency is equal to one quarter of the sampling rate for that stage.

The N samples of input signal S_1 are supplied in common to the inputs of highpass filter **40A** and lowpass filter **42A**. The corresponding N high frequency samples from highpass filter **40A** are downsampled by a factor of 2 by downsampler **41A** and the resulting N/2 samples supplied to the output as the highpass wavelet y_3. The N low frequency samples from lowpass filter **42A** are downsampled by a factor of 2 by downsampler **43A** and the resulting N/2 samples supplied to stage B where the same

procedure is repeated. In stage B, the N/2 higher frequency samples from highpass filter **40B** are downsampled by downsampler **41B** and the resulting N/4 samples supplied to the output as bandpass wavelet y_2. The other N/2 samples from lowpass filter **42B** are downsampled by downsampler **43B** and the resulting N/4 samples are supplied to the third stage C, in which highpass filter **40C** and downsampler **41C** process them in like manner to provide at the output N/8 samples as bandpass wavelet y_1. The other N/4 samples from lowpass filter **42C** are downsampled by downsampler **43C** to give N/8 samples and supplies them to the output as low-pass wavelet y_0.

It should be noted that, if the input signal segment comprises, for example, 1024 samples or data points, wavelets y_0 and y_1 comprise only 128 samples, wavelet y_2 comprises 256 samples and wavelet y_3 comprises 512 samples.

Instead of the octave-band structure of Figure 4A, a set of one lowpass, two bandpass filters and one highpass filter could be used, in parallel, with different downsampling rates.

Referring now to Figure 4B, in order to reconstruct the original input signal, the DWT wavelet signals are upsampled and passed through another set of lowpass and highpass filters, the operation being expressed as:

$$W_L(n,j) = \sum_k W_L(k,j+1)\, h'(n-2k) + \sum_l W_H(l,j+1)\, g'(n-2l) \qquad (2)$$

where $h'(n)$ and $g'(n)$ are, respectively, the lowpass and highpass synthesis filters corresponding to the mother wavelet. It is observed from equation 2 that j-th level DWT wavelet signals can be obtained from $(j+1)$-th level DWT coefficients.

Compactly supported wavelets are generally used in various applications. Table I lists a few orthonormal wavelet filter coefficients $h(n)$ that are popular in various applications as disclosed by I. Daubechies, in "Orthonormal bases of compactly supported wavelets", Comm. Pure Appl. Math, Vol. 41, pp. 906-966, 1988. These wavelets have the property of having the maximum number of vanishing moments for a given order, and are known as "Daubechies wavelets".

Coefficients	Wavelets	
	Daub-6	Daub-8
h(0)	0.332671	0.230378
h(1)	0.806892	0.714847
h(2)	0.459878	0.630881
h(3)	-0.135011	-0.027984
h(4)	-0.085441	-0.187035
h(5)	0.035226	0.030841
h(6)		0.032883
h(7)		-0.010597

Table I

An embodiment of the invention in which the higher sub-bands are not transmitted, and which uses discrete wavelet transforms for encoding a digital signal, will now be described with reference to Figure 5. In the transmitter/encoder 11' of Figure 5, the input signal S_1, is supplied to an input port **20** of analysis filter means comprising an octave-band filter bank **51** for applying Discrete Wavelet Transform as illustrated in Figure 4A to the signal S_1 to generate lowpass sub-band wavelet signal y_0, two bandpass sub-band wavelet signals, y_1 and y_2, and the highpass sub-band wavelet signal y_3. In this implementation, only sub-band wavelet signals y_0, y_1 and y_2 will be processed. Highpass sub-band wavelet signal y_3 is discarded. Interpolator means **52** interpolates sub-band wavelet signals y_0, y_1 and y_2 by factors 2M, 2M and M, respectively, where M is an integer, typically 8 to 24, such that the three sub-band wavelet signals (y_0, y_1 and y_2) have equal sample rates. Thus, within interpolator **52**, the sub-band wavelet signals y_0, y_1 and y_2 are upsampled by upsamplers 53_0, 53_1 and 53_2, respectively, which insert zero value samples at intervals between actual samples. The upsampled signals then are filtered by three Raise-Cosine filters 54_0, 54_1 and 54_2, respectively, which insert at each upsampled "zero" point a sample calculated from actual values of previous samples. The Raise-Cosine filters are preferred so as to minimize intersymbol interference. The three interpolated sub-band wavelet signals are supplied to double side-band (DSB) multi-carrier modulator **55** which uses them to

modulate three separate carrier signals f_0, f_1 and f_2, where $f_0 < f_1 < f_2$ provided by carrier generator **56**. The modulator **55** comprises multipliers **57**$_0$, **57**$_1$ and **57**$_2$ which multiply the carrier signals f_0, f_1 and f_2 by the three interpolated wavelet signals y^u_0, y^u_1 and y^u_2, respectively. The resulting three modulated carrier signals y'_0, y'_1 and y'_2 are added together by a summer **58** to form the encoded signal S_o for transmission by way of port **24** to transmission medium **12**.

At the corresponding decoder 13′ shown in Figure 6, the signal S'_o received at port **30** is supplied to each of three bandpass filters **61**$_0$, **61**$_1$ and **61**$_2$ which recover the modulated carrier signals y''_0, y''_1 and y''_2. The recovered modulated carrier signals y''_0, y''_1 and y''_2 are demodulated using multi-carrier double sideband (DSB) demodulator **62**. A carrier generator **63** generates carrier signals having frequencies f_0, f_1 and f_2, which are supplied to multipliers **64**$_0$, **64**$_1$ and **64**$_2$ within the demodulator **62** and which multiply the carrier signals f_0, f_1 and f_2 by the recovered modulated carrier signals y''_0, y''_1 and y''_2, respectively. The DSB demodulator **62** comprises lowpass filters **65**$_0$, **65**$_1$, and **65**$_2$ for filtering the outputs of the multipliers **64**$_0$, **64**$_1$ and **64**$_2$, respectively, as is usual in a DSB demodulator.

The demodulated signals from the filters **65**$_0$, **65**$_1$ and **65**$_2$ are decimated by 2M, 2M and M, respectively, by decimators **66**$_0$, **66**$_1$ and **66**$_2$ of a decimator unit **66** and the resulting recovered sub-band signals y^*_0, y^*_1 and y^*_2 each supplied to a corresponding one of four inputs of a synthesis filter bank **67** which applies to them an Inverse Discrete Wavelet Transform (IDWT) as illustrated in Figure 4B to recover the signal S'_i which corresponds to the input signal S_i. The highpass sub-band wavelet signal y_3, which was not transmitted, is replaced by a "zero" signal at the corresponding "highest" frequency input **68** of the synthesis filter bank **67**. The resulting output signal S'_i from the synthesis filter bank **67** is the decoder output signal supplied via output port **34**, and is a close approximation of the input signal S_i supplied to the encoder **11′** of Figure 5.

The bandwidth of the transmitted signal S_o is wider than that of the original signal S_i because each sub-band has upper and lower sidebands. A bandwidth reduction can be achieved by using Single Sideband (SSB) modulation. To do so, the encoder **11′** of Figure 5 would be modified by replacing each of the multipliers **57**$_0$, **57**$_1$ and **57**$_2$ by a SSB modulator. Figures 7A, 7B and 7C illustrate operation of the encoder using very simplified signals and, for convenience of illustration, SSB modulation.

Figure 7A shows the frequency spectrum of a much-simplified input signal S_i

occupying a bandwidth BW centered at frequency f_c. As shown in Figure 7B, after analysis filtering and interpolation, the input signal S_i has been partitioned into three interpolated sub-band signals, y''_0, y''_1 and y''_2. It should be noted that, for complex input signals, the sub-band signals y_0, y_1 and y_2 prior to interpolation have a very wide spectrum. After upsampling and filtering by the interpolator 52 (Figure 5), sub-band signals y''_0, y''_1 and y''_2 each have a frequency spectrum that is much narrower than the frequency spectrum of the original signal S_i.

Following modulation by the DSB multi-carrier modulation means 55, the bandwidths BW_0, BW_1, and BW_2 of the corresponding modulated carriers y'_0, y'_1 and y'_2 are determined by the sampling rate of the input signal S_i. The total bandwidth $BW_0 + BW_1 + BW_2 + 2G$ may be greater than the bandwidth BW if all sub-bands are used, but may be less if only two are used. The output signal S_o from the summing means 58 has a spectrum which, as shown in Figure 7C, has three lobes, namely a lower frequency lobe centered at frequency f_0, a middle frequency lobe centered at frequency f_1 and an upper frequency lobe centered at frequency f_2. The three lobes are separated from each other by two guard bands G to avoid interference and ensure that each carries information for its own sub-band only.

Simplified versions of the input signal S_i, sub-band wavelet signals y_0, y_1, y_2 and y_3, sub-band wavelet modulated carriers y'_0, y'_1 and y'_2, and the transmitted signal S_o, which are similar in the encoders of Figures 2 and 5, are shown in Figures 8 - 10. Figure 8 shows the simplified input signal S_o, (which is not the same as that illustrated in Figure 7A). Figures 9A, 9B, 9C and 9D illustrate the sub-band wavelet signals y_0, y_1, y_2 and y_3 obtained by DWT processing of the input signal S_i. Figures 10A, 10B and 10C illustrate the corresponding modulated carrier signals y'_0, y'_1 and y'_2 obtained by modulating the carrier signals f_0, f_1 and f_2 with the sub-band wavelet signals y_0, y_1 and y_2, respectively. Because the waveform of the simplified input signal is so smooth, the wavelet signal y_2 is interpolated by a factor of 2 only, and the wavelet signal y_0 and y_1 by a factor of 4 only. This is, of course, for illustration only; in practice the interpolator may typically range from 1:8 to 1:24. Figure 11 shows the encoded signal S_o and Figure 12 shows its frequency spectrum which comprises the spectrum components of y'_0, y'_1 and y'_2 centered at frequencies of 1000 Hertz, 3000 Hertz and 5000 Hertz, respectively. for a message rate of 750 Hertz. The asymmetric distribution of transmission power between the lower and high frequency carriers should be noted. It should be appreciated

that these simplified signals are for illustration only and that real signals would be much more complex.

Figures 13A, 13B and 13C illustrate the recovered modulated carrier signals y''_0, y''_1, and y''_2, and Figures 14A, 14B and 14C illustrate the recovered sub-band wavelet signals y^*_0, y^*_1, and y^*_2. Finally, Figure 15 illustrates the reconstructed signal S'_i which can be seen to be a close approximation of the input signal S_i shown in Figure 8.

In the above-described embodiment, the highpass sub-band signal y_3 is not used, on the grounds that it probably contains negligible energy. If it has significant energy, however, it could be used, and the encoder and decoder modified appropriately.

While similar implementations using more than two sub-bands and carriers are possible, and might be desirable in some circumstances, for most applications, and especially communication of digital signals via twisted wire subscriber loops, they would be considered complex without significant improvement in performance.

It should be appreciated that other kinds of modulation might be used to modulate the sub-band signals, for example, narrow-band frequency modulation, and so on.

It should also be appreciated that the signal source **10** and encoder **11** could be parts of a transmitter having other signal processing means. Likewise, the decoder **13** and signal destination **14** could be parts of a corresponding receiver.

Although the above-described embodiments of the invention use three or more of the sub-band signals, it is envisaged that other applications, such as deep space communications, might use only one or two of the wavelets.

INDUSTRIAL APPLICABILITY

An advantage of embodiments of the present invention, which use sub-band signals to modulate carriers, is that transmission is reliable because the impairment of one sub-band in the system would cause the transmission system to degrade only gently. Also, the decoder bandpass filters can be easily designed because there are only a few frequency bands used. Moreover, in applications involving data transmission, data synchronization and clock recovery can be easily achieved in the decoder.

It should be noted that the present invention is not limited to transmission systems but could be used for other purposes to maintain signal integrity despite noise and attenuation. For example, it might be used in recording of the signal on a compact disc or other storage medium. The storage medium can therefore be equated with the

transmission medium 12 in Figure 1. It should be appreciated that the encoders and decoders described herein would probably be implemented by a suitably programmed digital signal processor or as a custom integrated circuit.

Although embodiments of the invention have been described and illustrated in detail, it is to be clearly understood that the same are by way of illustration and example only and not to be taken by way of the limitation, the spirit and scope of the present invention being limited only by the appended claims.

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CLAIMS

1. Apparatus for encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, characterized by:

an encoder (11) comprising:

(i) analysis filter means (21;51) for analyzing the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of frequencies;

(ii) means (25;56) for providing a plurality of carrier signals having different frequencies;

(iii) modulation means (22;55) for using at least some of the sub-band signals each to modulate a respective one of the plurality of carrier signals; and

(iv) combining means (23;58) for combining the modulated carrier signals to form the encoded signal for transmission or storage;

and a decoder (13) comprising:

(v) filter means ($31;61_0$, 61_1, 61_2) for extracting the plurality of modulated carrier signals from the received or recorded encoded signal;

(vi) means (35;63) for providing a plurality of carrier signals corresponding to those of said encoder;

(vii) demodulation means (32;62) for using at least some of the carrier signals to demodulate the extracted modulated carrier signals to extract the corresponding plurality of sub-band signals; and

(vii) synthesis filter means (33;67) complementary to said analysis filter means for processing the plurality of extracted sub-band signals to produce a decoded signal corresponding to the input signal.

2. Apparatus as claimed in claim 1, characterized in that the analysis filter means comprises a uniform filter bank for producing sub-bands each having the same bandwidth.

3. Apparatus as claimed in claim 1, characterized in that the analysis filter means comprises a multiresolution filter bank for providing sub-bands having different bandwidths.

4. Apparatus as claimed in claim 3, characterized in that the multiresolution filter bank comprises an octave band filter bank (40A/B/C/D, 41A/B/C/D, 42A/B/C/D, 43A/B/C/D) implementing Discrete Wavelet Transform (DWT) to produce a plurality of wavelets as said plurality of sub-band signals, and the synthesis filter means comprises an octave band filter bank (Fig. 4B) implementing a corresponding Inverse Discrete Wavelet Transform.

5. Apparatus as claimed in claim 1, further characterized in that, for encoding digital input signals, the encoder (11') further comprises interpolation means (52) for providing, for each sub-band signal, interpolated values between adjacent actual values of the sub-band signal, and the decoder (13') further comprises a decimator (66) for decimating the extracted sub-band signals to remove interpolated values before application to the synthesis filter means, the interpolation rates being such that the resulting interpolated sub-band signals all have the same rate.

6. Apparatus as claimed in claim 5, characterized in that the interpolation means (52) comprises an upsampler (53_0, 53_1, 53_2) for interpolating intervals between actual values and means (54_0, 54_1, 54_2) for determining values between the actual samples and inserting such determined values at the appropriate intervals.

7. Apparatus as claimed in claim 6, characterized in that the determining means (54_0, 54_1, 54_2) comprises Raise Cosine filter means.

8. Apparatus as claimed in claim 1, characterized in that in the encoder (11), the modulation means (22;55) uses fewer than the total number of said plurality of sub-band signals to modulate said carrier signals and, in the decoder (13), the synthesis filter means (33;67) substitutes zero level signals for the unused sub-band signals.

9. An encoder for use in the apparatus according to claim 1, and for encoding an input signal for transmission or storage, characterized by:

(i) analysis filter means (21;51) for analyzing the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of frequencies;

(ii) means (25;56) for providing a plurality of carrier signals having different frequencies;

(iii) modulation means (22;55) for using at least some of the sub-band signals each to modulate a respective one of the plurality of carrier signals; and

(iv) combining means (23;58) for combining the modulated carrier signals to form the encoded signal for transmission or storage.

10. An encoder as claimed in claim 9, for use in the apparatus of claim 2 and characterized in that the analysis filter means comprises a uniform filter bank for producing sub-bands each having the same bandwidth.

11. An encoder as claimed in claim 9, for use in the apparatus of claim 3 and characterized in that the analysis filter means comprises a multiresolution filter bank for providing sub-bands having different bandwidths.

12. An encoder as claimed in claim 11, for use in the apparatus of claim 4 and characterized in that the multiresolution filter bank comprises an octave band filter bank (40A/B/C/D, 41A/B/C/D, 42A/B/C/D, 43A/B/C/D) implementing Discrete Wavelet Transform (DWT) to produce a plurality of wavelets as said plurality of sub-band signals.

13. An encoder as claimed in claim 9, for use in the apparatus of claim 5 and characterized in that, for encoding digital input signals, the encoder further comprises interpolation means (52) for providing, for each sub-band signal, interpolated values between adjacent actual values of the sub-band signal, for use with a decoder further comprising a decimator (66) for decimating the extracted sub-band signals to remove interpolated values before application to the synthesis filter means, the interpolation rate being such that the resulting interpolated sub-band signals all have the same rate.

14. An encoder as claimed in claim 13, for use in the apparatus of claim 6 and characterized in that the interpolation means (52) comprises an upsampler (53_0, 53_1, 53_2) for interpolating intervals between actual values and filter means (54_0, 54_1, 54_2) for determining values between the actual samples and inserting such determined values at

the appropriate intervals.

15. An encoder as claimed in claim 14, for use in the apparatus of claim 7 and characterized in that the determining means (54_0, 54_1, 54_2) comprises Raise Cosine filter means.

16. An encoder as claimed in claim 9, for use in the apparatus of claim 8 and characterized in that the modulation means (22;55) uses fewer than the total number of said plurality of sub-band signals to modulate said carrier signals.

17. A decoder (13) for decoding an encoded signal encoded by analysis filtering an input signal to produce a plurality of sub-band signals, using at least some of the sub-band signals to modulate a plurality of carrier signals, and combining the modulated carrier signals to form the encoded signal, the decoder being characterized by:

filter means (31;61_0, 61_1, 61_2) for recovering the plurality of modulated carrier signals from the encoded signal;

means (35;63) for providing a plurality of carrier signals corresponding to those used to encode the encoded signal;

demodulation means (32;62) for using at least some of the carrier signals to demodulate the recovered modulated carrier signals to extract the corresponding plurality of sub-band signals; and

synthesis filter means (33;67) complementary to the analysis filter means used to encode the encoded signal for processing the plurality of extracted sub-band signals to produce a decoded signal.

18. A decoder as claimed in claim 17, for decoding encoded signals derived by encoding a digital input signal and interpolating values between adjacent actual values of the sub-band signals, the interpolation rate being such that the resulting interpolated sub-band signals all have the same rate, the decoder further characterized by a decimator (66) for decimating the extracted sub-band signals to remove interpolated values before application to the synthesis filter means.

19. A decoder as claimed in claim 17 or 18, characterized in that, for decoding an

encoded signal encoded using fewer than the total number of sub-bands generated by the analysis filter means, the synthesis filter means (33;67) substitutes zero-level signals for the unused sub-band signals.

20. A method of encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, <u>characterized in that</u> the encoding of the input signal comprises the steps of:

(i) using analysis filter means, analyzing the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of frequencies;

(ii) providing a plurality of carrier signals having different frequencies;

(iii) using at least some of the sub-band signals each to modulate a respective one of the plurality of carrier signals; and

(iv) combining the modulated carrier signals to form the encoded signal for transmission or storage;

and the decoding of the encoded signal comprises the steps of:

(v) extracting the plurality of modulated carrier signals from the received or recorded encoded signal;

(vi) providing a plurality of carrier signals corresponding to those used in said encoding;

(vii) using at least some of the carrier signals to demodulate the extracted modulated carrier signals to extract the corresponding plurality of sub-band signals; and

(vii) using synthesis filter means complementary to said analysis filter means, processing the plurality of extracted sub-band signals to produce a decoded signal corresponding to the input signal.

21. A method as claimed in claim 20, <u>characterized in that</u> the analysis filtering uses a uniform filter bank to produce sub-bands each having the same bandwidth.

22. A method as claimed in claim 20, <u>characterized in that</u> the analysis filtering uses a multiresolution filter bank to provide sub-bands having different bandwidths.

23. A method as claimed in claim 22, characterized in that the analysis filtering uses an octave band filter bank implementing Discrete Wavelet Transform (DWT) to produce a plurality of wavelets as said plurality of sub-band signals, and the synthesis filtering uses an octave band filter bank implementing a corresponding Inverse Discrete Wavelet Transform.

24. A method as claimed in claim 20, further characterized in that, for encoding digital input signals, the encoding further comprises the step of interpolating each sub-band signal by inserting interpolated values between adjacent actual values of the sub-band signal, and the decoding further comprises decimating the extracted sub-band signals to remove interpolated values before synthesis filtering, the interpolation rates being such that the resulting interpolated sub-band signals all have the same rate.

25. A method as claimed in claim 24, characterized in that the interpolation includes the step of upsampling the sub-band signal to interpolate intervals between actual values, using previous actual to determine interpolated values and inserting such determined interpolated values at the appropriate intervals.

26. A method as claimed in claim 25, characterized in that the determination of the interpolated values uses Raise Cosine filtering.

27. A method as claimed in claim 20, wherein the modulation uses fewer than the total number of said plurality of sub-band signals to modulate said carrier signals and, in the processing by the synthesis filter means, the unused sub-band signals are replaced by zero level signals.

28. A method of encoding an input signal for transmission or storage, characterized by the steps of:

(i) using analysis filter to analyze the input signal into a plurality of sub-band signals, each sub-band centered at a respective one of a corresponding plurality of frequencies;

(ii) providing a plurality of carrier signals having different frequencies;

(iii) using at least some of the sub-band signals each to modulate a respective

one of the plurality of carrier signals; and

(iv) combining the modulated carrier signals to form the encoded signal for transmission or storage.

29. An encoding method as claimed in claim 28, characterized in that the analysis filtering uses a uniform filter bank for producing sub-bands each having the same bandwidth.

30. An encoding method as claimed in claim 28, characterized in that the analysis filtering uses a multiresolution filter bank for providing sub-bands having different bandwidths.

31. An encoding method as claimed in claim 29, characterized in that the analysis filtering uses an octave band filter bank implementing Discrete Wavelet Transform (DWT) to produce a plurality of wavelets as said plurality of sub-band signals.

32. An encoding method as claimed in claim 28, characterized in that, for encoding digital input signals, the encoding further comprises the step of interpolating each sub-band signal to provide interpolated values between adjacent actual values of the sub-band signal, the interpolation rate being such that the resulting interpolated sub-band signals all have the same rate.

33. An encoding method as claimed in claim 32, characterized in that the interpolation comprises the step of upsampling the sub-band signal to establish intervals between actual values, determining values between the actual samples and inserting such determined values at the appropriate intervals.

34. An encoding method as claimed in claim 33, characterized in that the values are determined using Raise Cosine filtering.

35. A method of decoding an encoded signal encoded by analysis filtering an input signal to produce a plurality of sub-band signals, using at least some of the sub-band signals to modulate a plurality of carrier signals, and combining the modulated carrier

signals to form the encoded signal, the decoding method being characterized by the steps of:

recovering the plurality of modulated carrier signals from the encoded signal;

providing a plurality of carrier signals corresponding to those used to encode the encoded signal;

using at least some of the carrier signals to demodulate the recovered modulated carrier signals to extract the corresponding plurality of sub-band signals; and

using synthesis filter means complementary to the analysis filter means used to encode the encoded signal, processing the plurality of extracted sub-band signals to produce a decoded signal.

36. A decoding method as claimed in claim 35, for decoding encoded signals derived by encoding a digital input signal and interpolating values between adjacent actual values of the sub-band signals, the interpolation rate being such that the resulting interpolated sub-band signals all have the same rate, the decoding method further characterized by the step of decimating the extracted sub-band signals to remove interpolated values before application to the synthesis filter means.

37. A decoding method as claimed in claim 35 or 36, characterized in that, for decoding an encoded signal encoded using fewer than the total number of sub-bands generated by the analysis filter means, the processing step by synthesis filter means (33;67) substitutes zero-level signals for the unused sub-band signals.

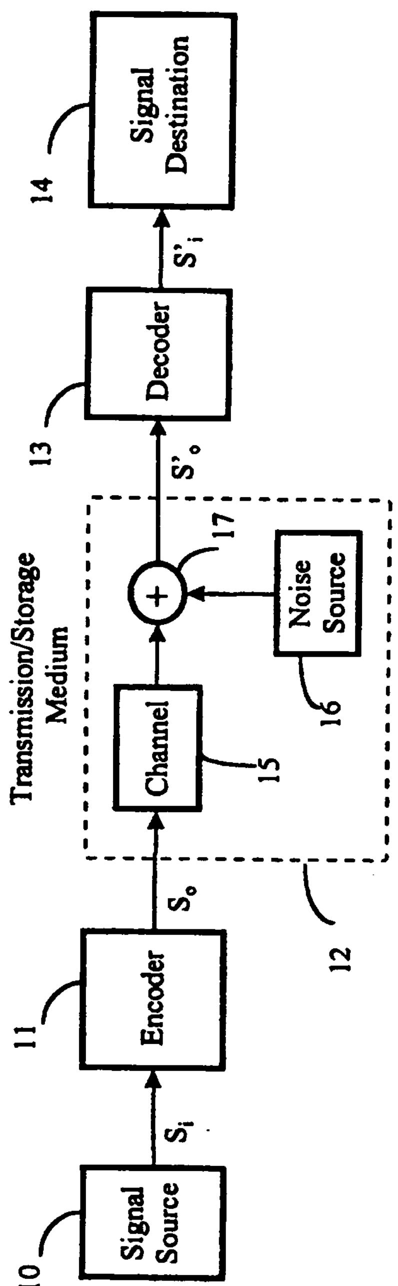

Signal Source
10
S_i
Encoder
11
S_o
Transmission/Storage Medium
12
Channel
15
+
17
Noise Source
16
S'_o
Decoder
13
S'_i
Signal Destination
14

FIG. 1

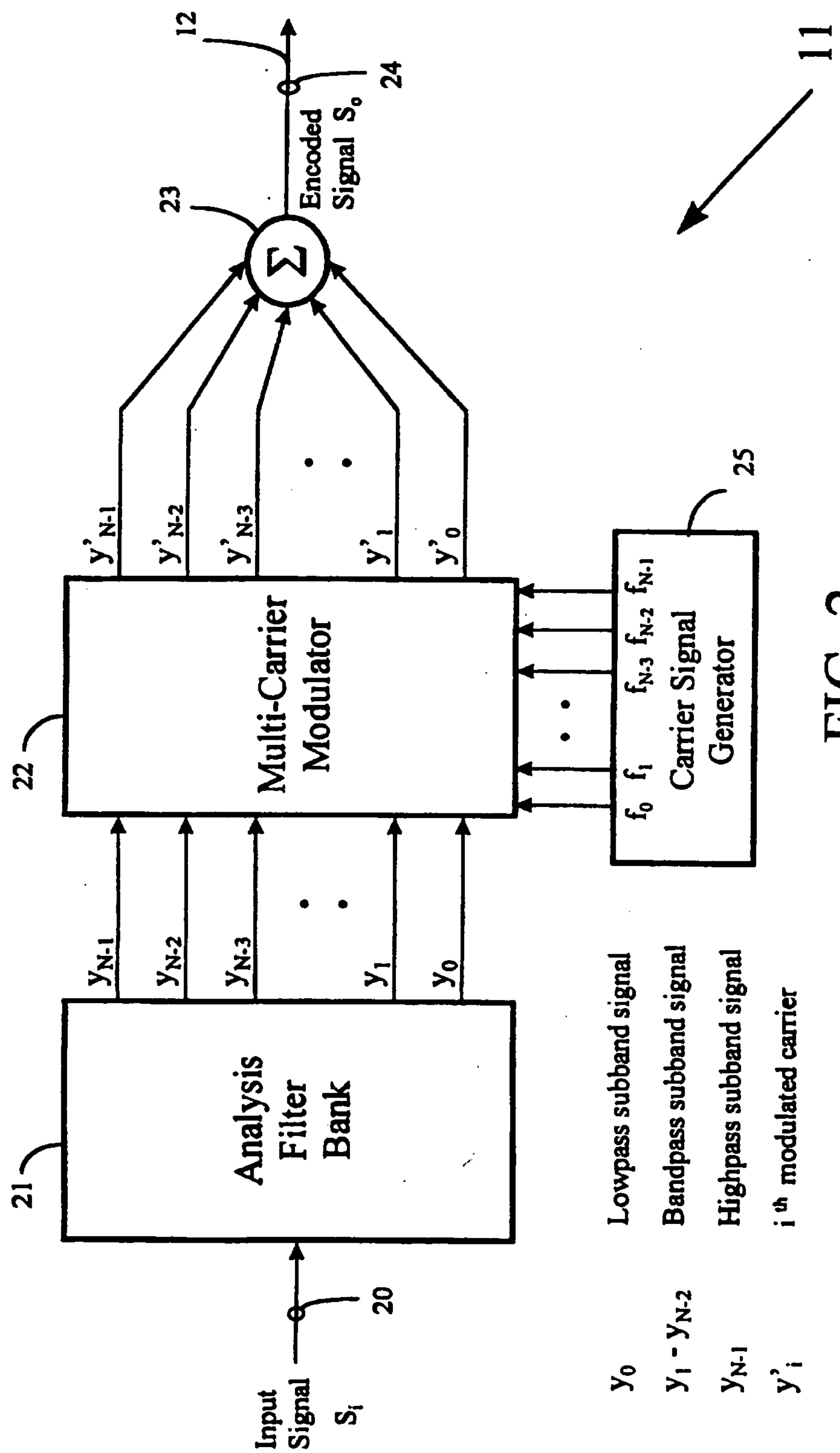

Input Signal Si
20
21
Analysis Filter Bank
yN-1
yN-2
yN-3
y1
y0
22
Multi-Carrier Modulator
y'N-1
y'N-2
y'N-3
y'1
y'0
f0 f1 fN-3 fN-2 fN-1
Carrier Signal Generator
25
23
Σ
Encoded Signal So
24
12
11
y0 Lowpass subband signal
y1 - yN-2 Bandpass subband signal
yN-1 Highpass subband signal
y'i i th modulated carrier

FIG. 2

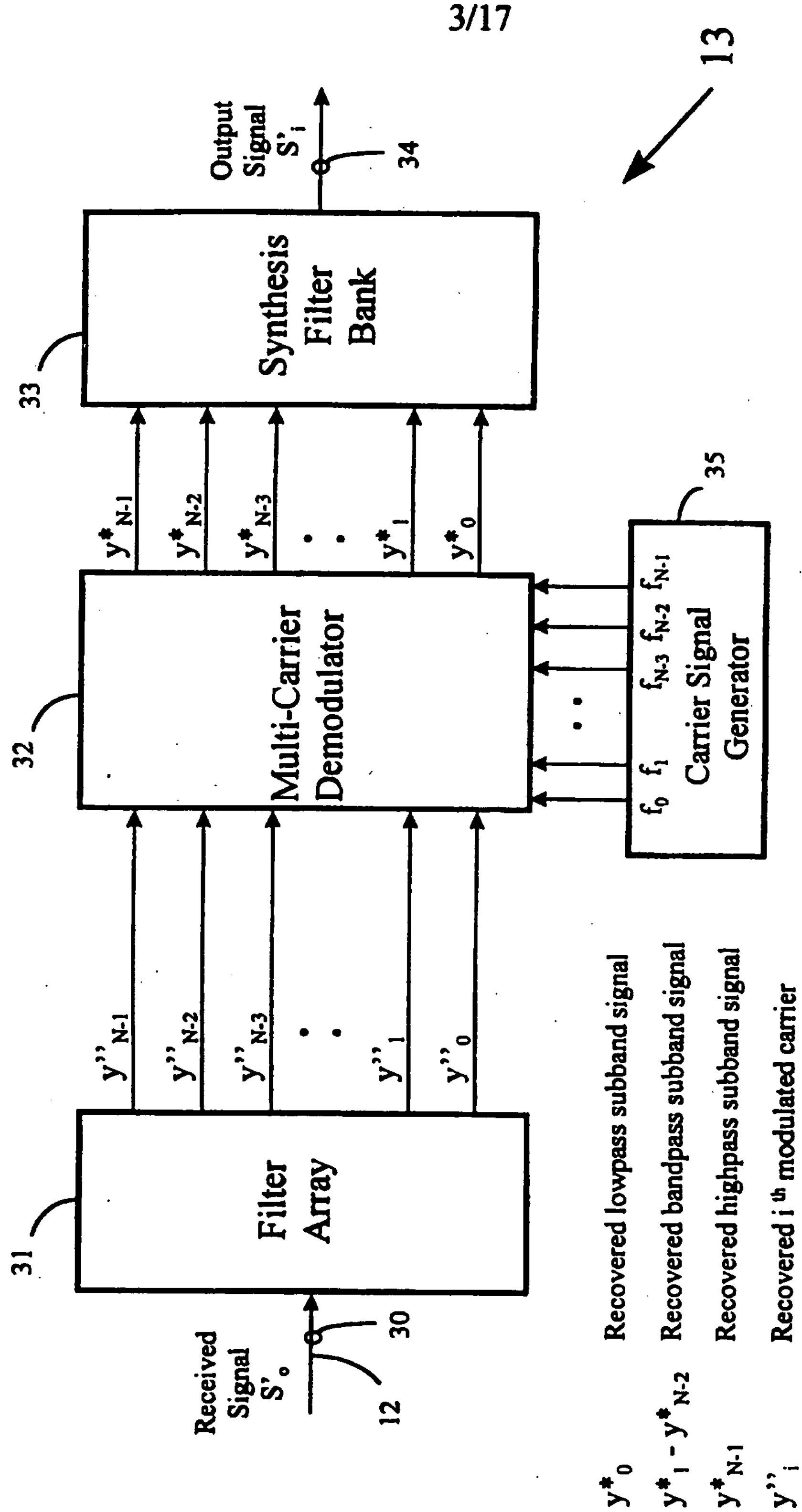

13
Received Signal S'_o
12
30
31
Filter Array
y''_{N-1}
y''_{N-2}
y''_{N-3}
y''_1
y''_0
32
Multi-Carrier Demodulator
y^*_{N-1}
y^*_{N-2}
y^*_{N-3}
y^*_1
y^*_0
35
f_0 f_1 f_{N-3} f_{N-2} f_{N-1}
Carrier Signal Generator
33
Synthesis Filter Bank
34
Output Signal S'_i
y^*_0 Recovered lowpass subband signal
$y^*_1 - y^*_{N-2}$ Recovered bandpass subband signal
y^*_{N-1} Recovered highpass subband signal
y''_i Recovered i th modulated carrier

FIG. 3

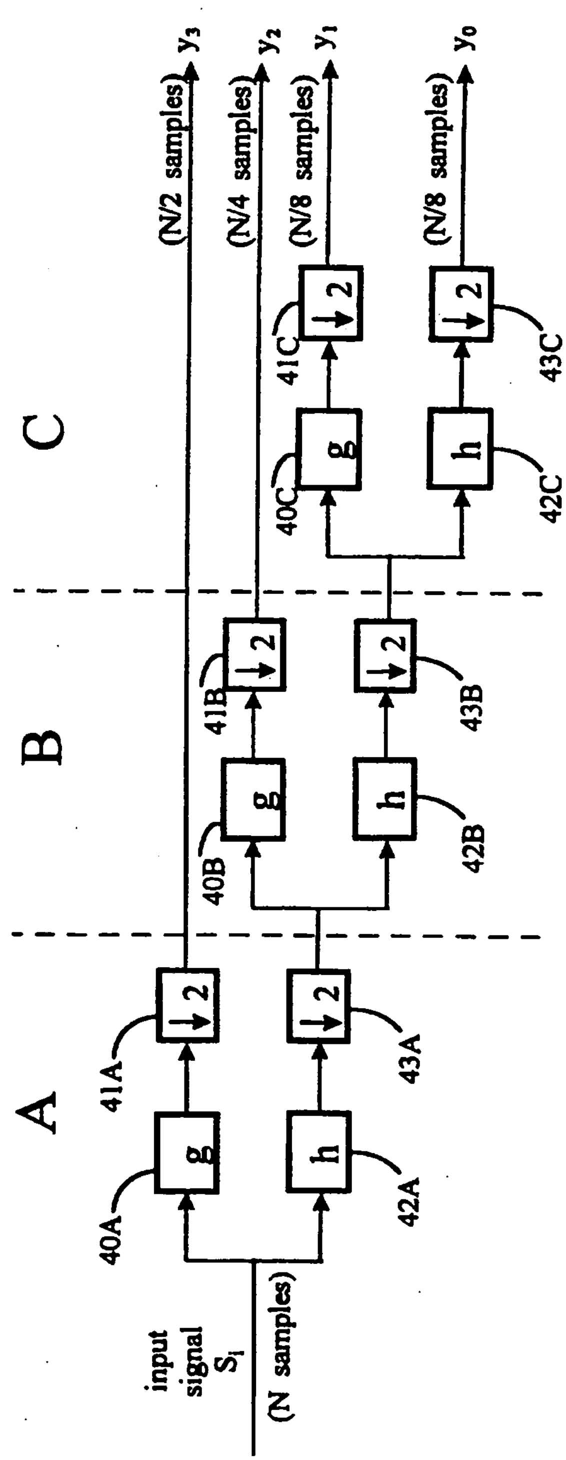

A
B
C
input signal S_I
(N samples)
40A
41A
42A
43A
40B
41B
42B
43B
40C
41C
42C
43C
g
h
↓2
(N/2 samples)
(N/4 samples)
(N/8 samples)
(N/8 samples)
y_3
y_2
y_1
y_0

FIG. 4A

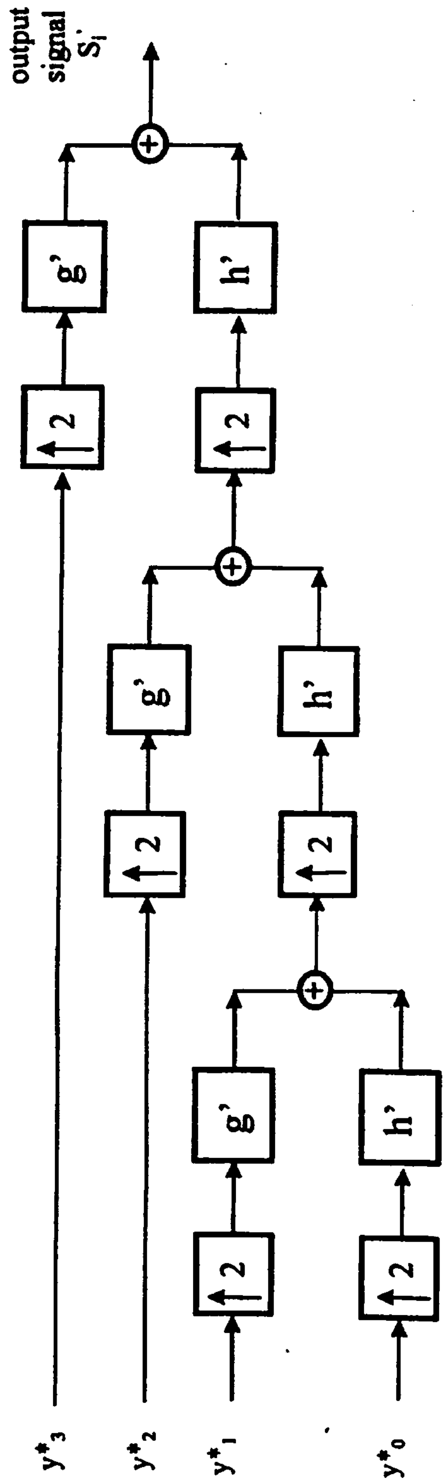


FIG. 4B

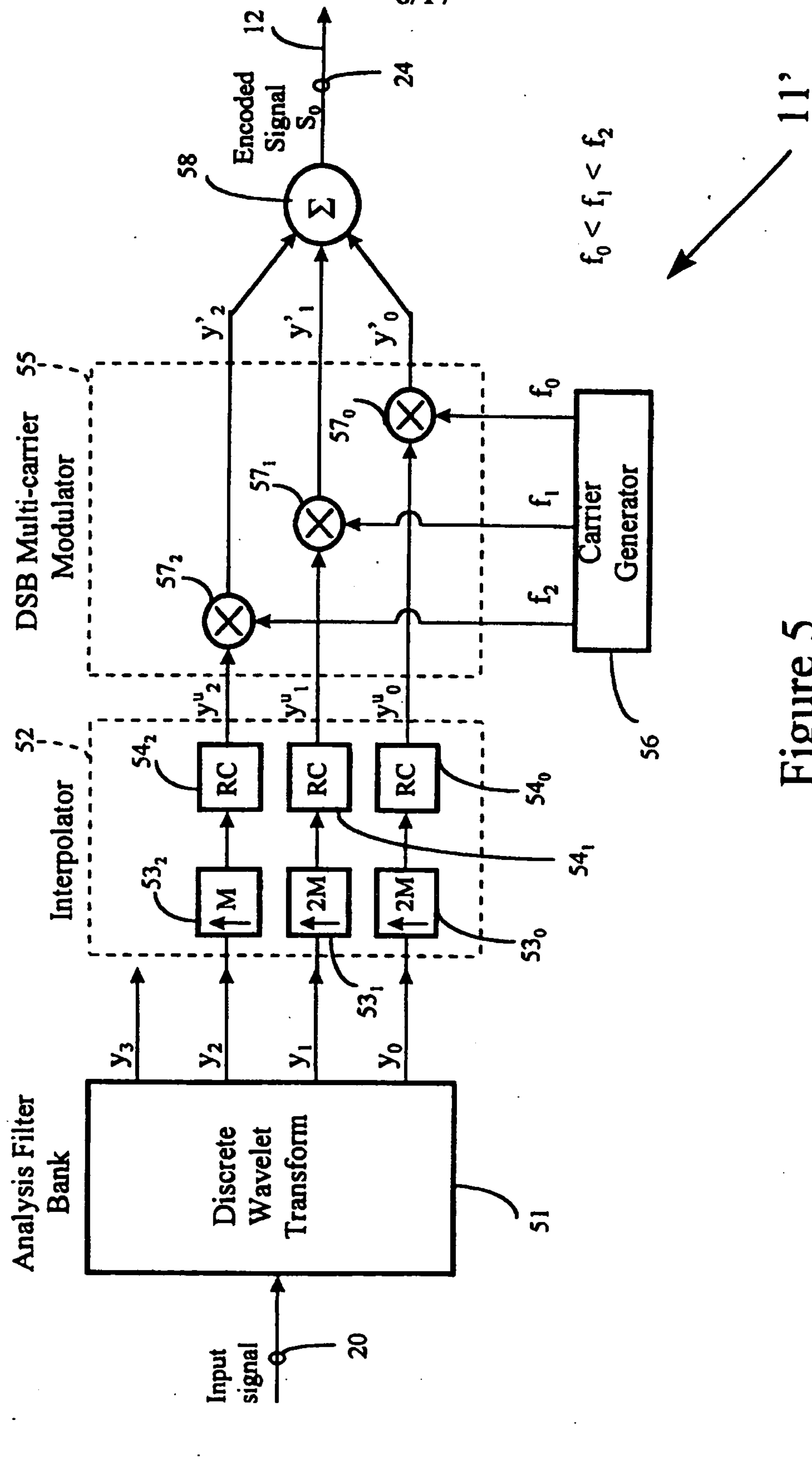

Input signal
20
Analysis Filter Bank
Discrete Wavelet Transform
51
y_3
y_2
y_1
y_0
Interpolator
52
53_2
53_1
53_0
M
2M
2M
RC
RC
RC
54_2
54_1
54_0
y^u_2
y^u_1
y^u_0
DSB Multi-carrier Modulator
55
57_2
57_1
57_0
f_2
f_1
f_0
Carrier Generator
56
y'_2
y'_1
y'_0
58
Σ
Encoded Signal S_0
24
12
$f_0 < f_1 < f_2$
11'

Figure 5

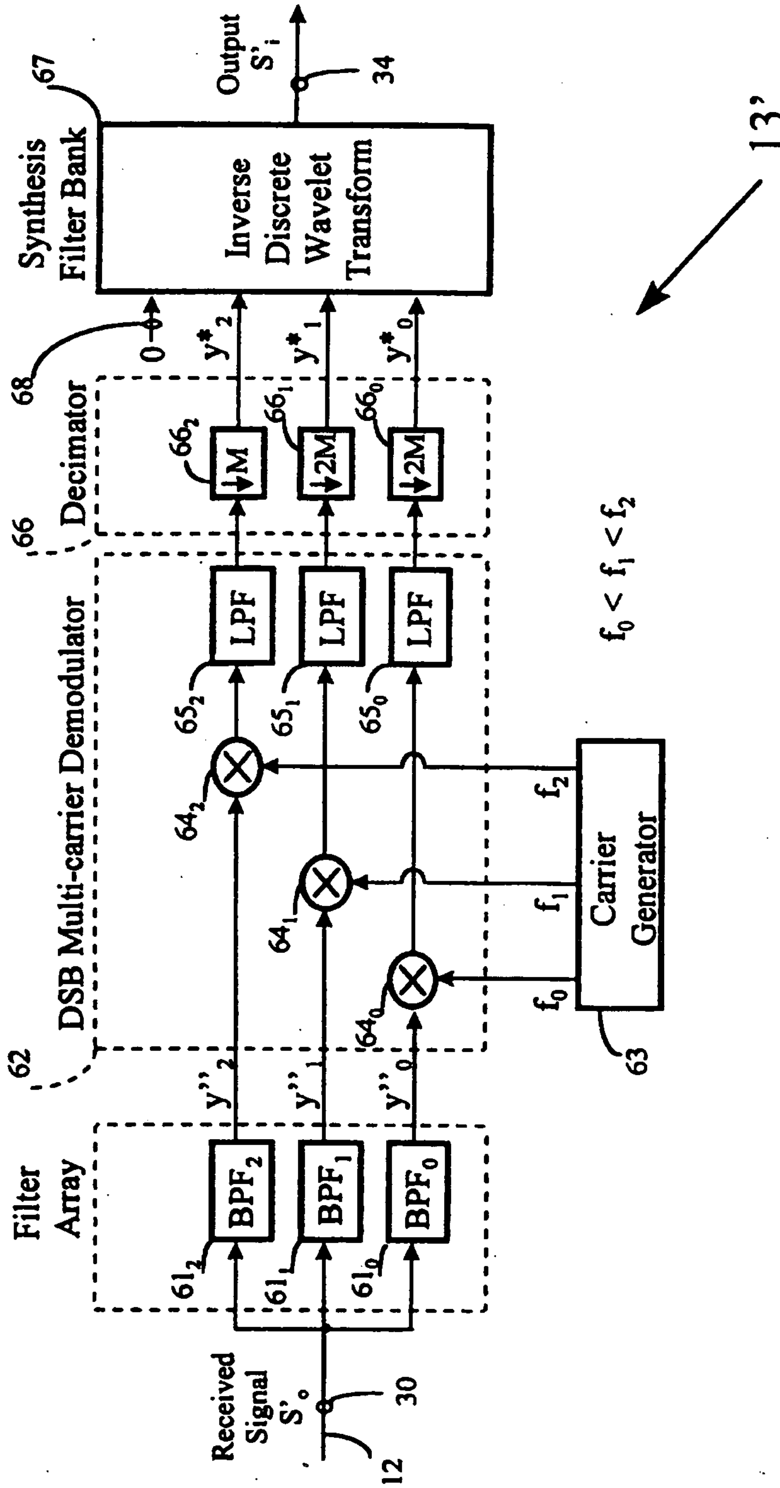
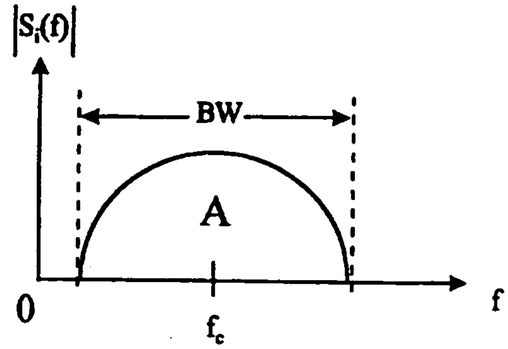

Received Signal S'o
12
30
Filter Array
BPF2
BPF1
BPF0
612
611
610
y''2
y''1
y''0
62
DSB Multi-carrier Demodulator
642
641
640
f0
f1
f2
Carrier Generator
63
652
651
650
LPF
LPF
LPF
f0 < f1 < f2
66
Decimator
662
661
660
↓M
↓2M
↓2M
68
0
y*2
y*1
y*0
Synthesis Filter Bank
67
Inverse Discrete Wavelet Transform
Output S'i
34
13'

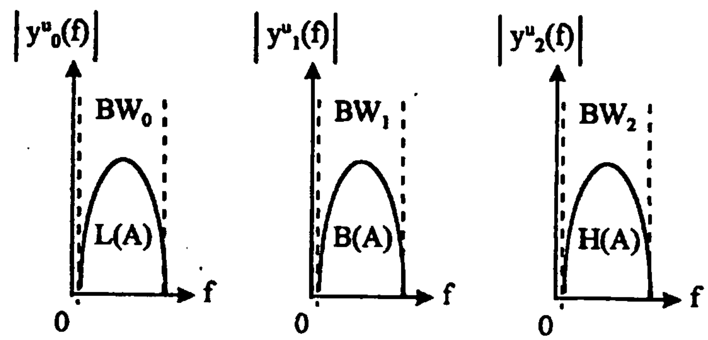
FIG. 6

FIG. 7A



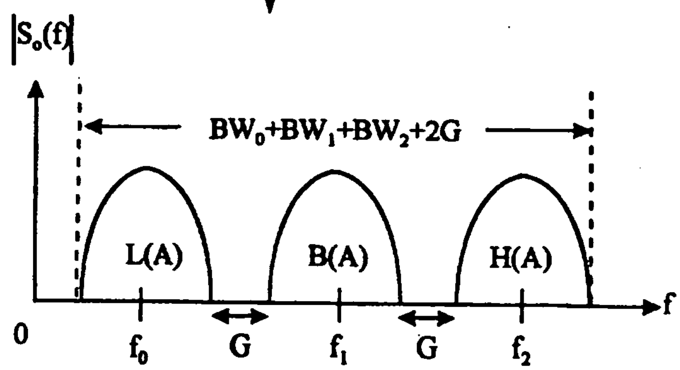
Analysis Filter +
Upsample + RC filter

FIG. 7B



Multi-carrier SSB

FIG. 7C



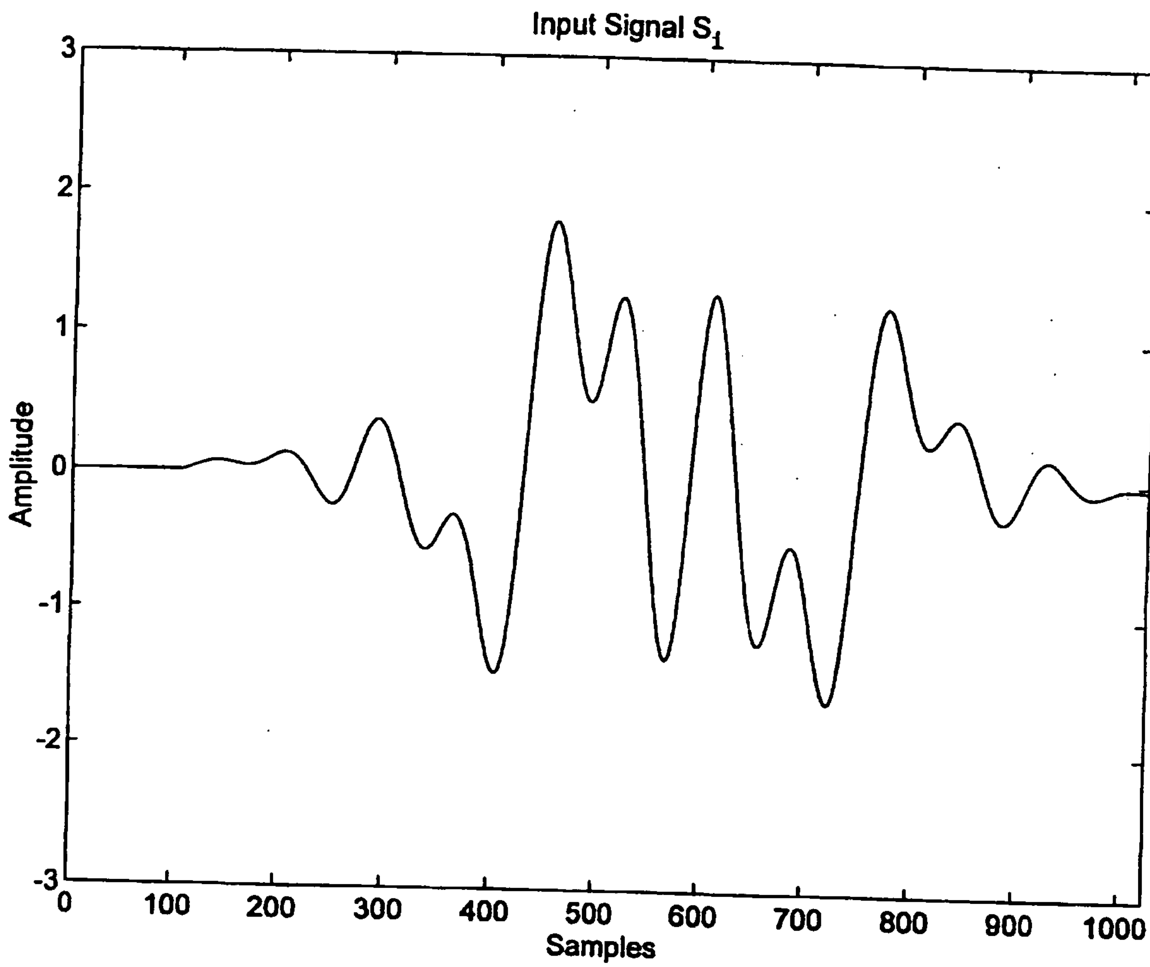

Input Signal S1
3
2
1
0
-1
-2
-3
Amplitude
0
100
200
300
400
500
600
700
800
900
1000
Samples

FIG. 8

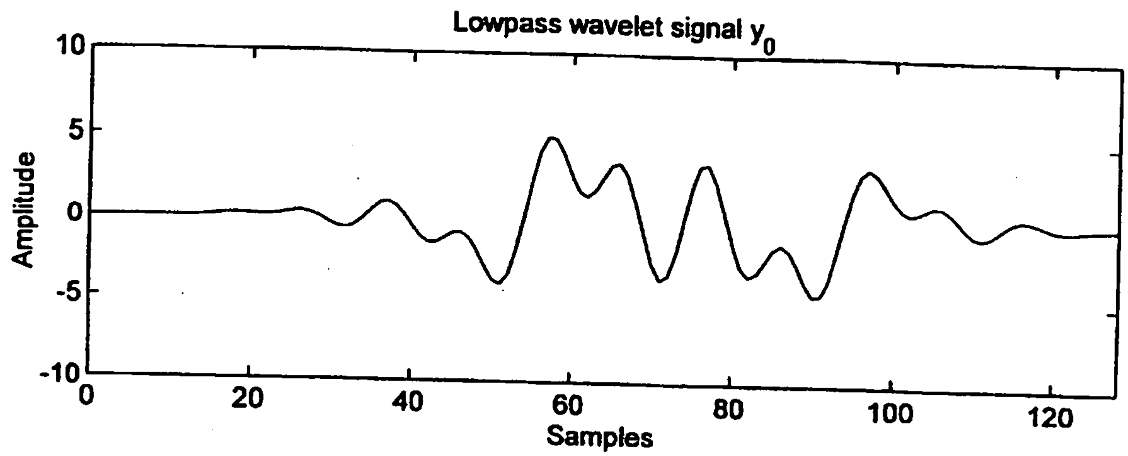

Lowpass wavelet signal y_0
10
5
0
-5
-10
Amplitude
0
20
40
60
80
100
120
Samples

FIG. 9A

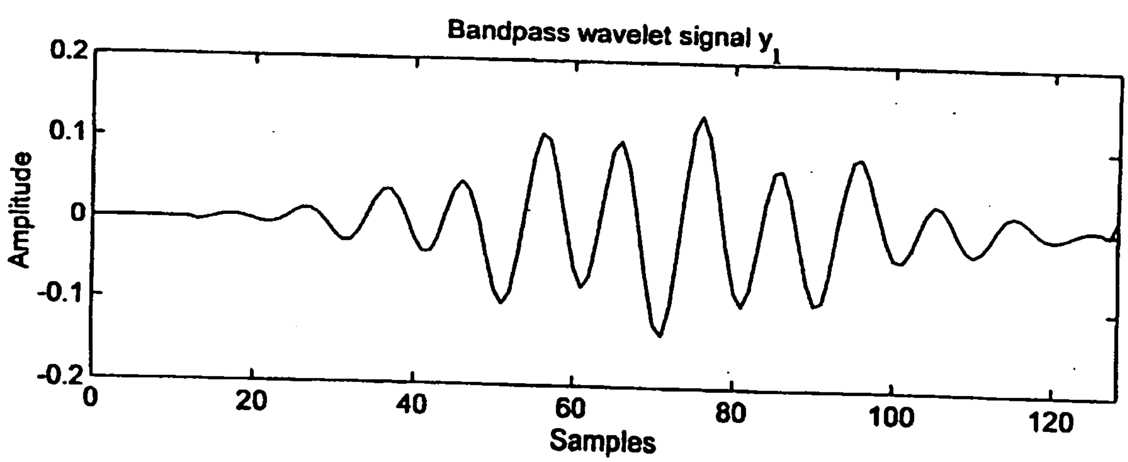

Bandpass wavelet signal y_1
0.2
0.1
0
-0.1
-0.2
Amplitude
0
20
40
60
80
100
120
Samples

FIG. 9B

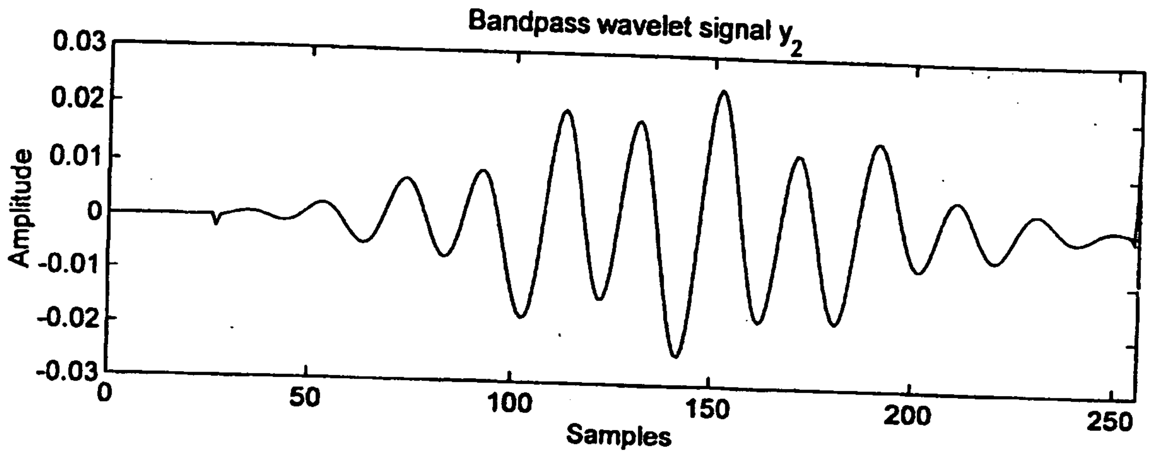

Bandpass wavelet signal y_2
0.03
0.02
0.01
0
-0.01
-0.02
-0.03
Amplitude
0
50
100
150
200
250
Samples

FIG. 9C

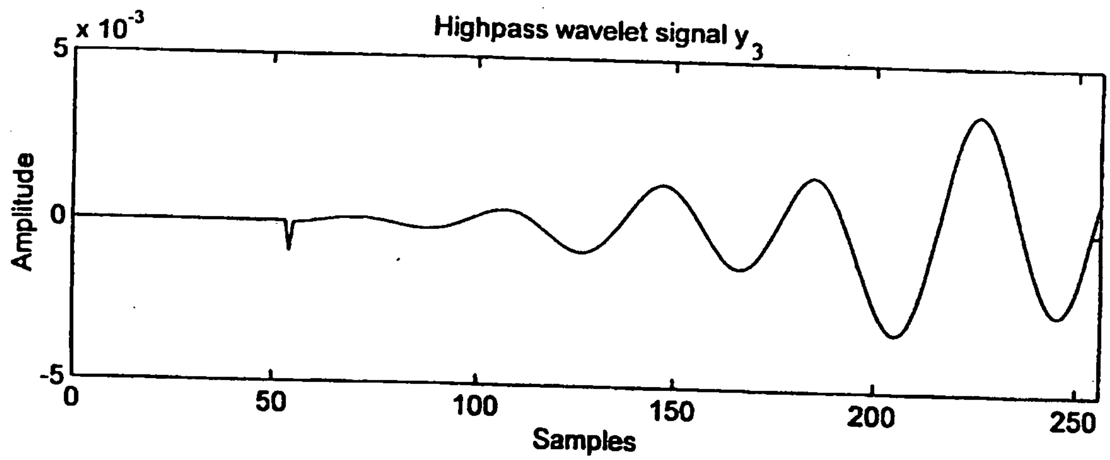

x 10^{-3}
Highpass wavelet signal y_3
5
0
-5
Amplitude
0
50
100
150
200
250
Samples

FIG. 9D

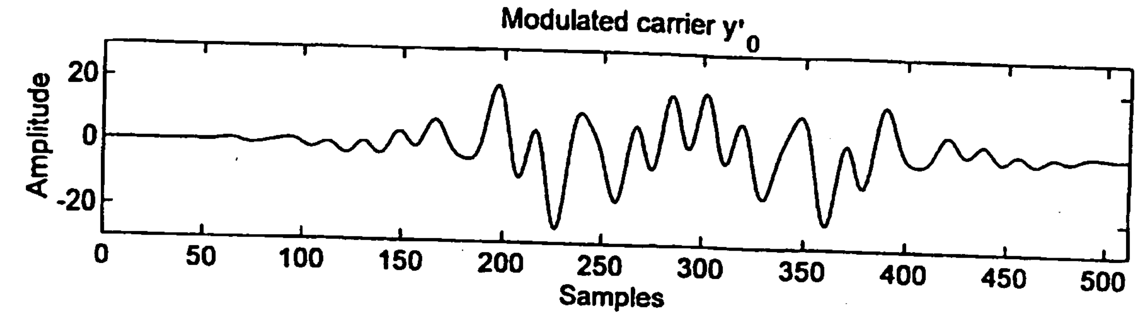


FIG. 10A

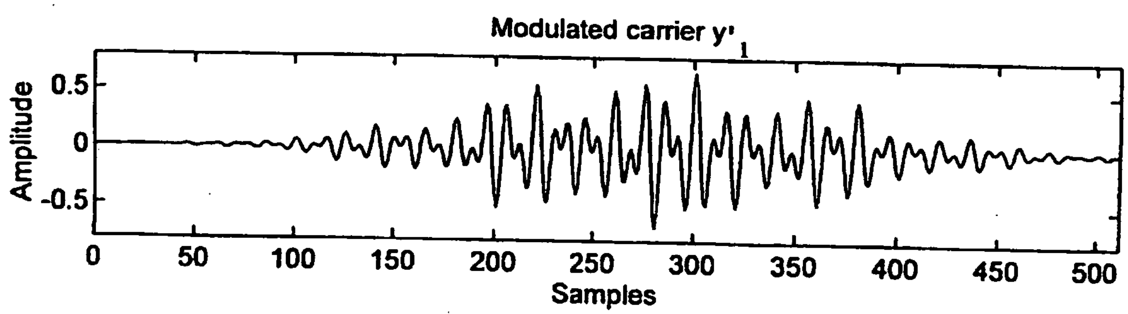


FIG. 10B

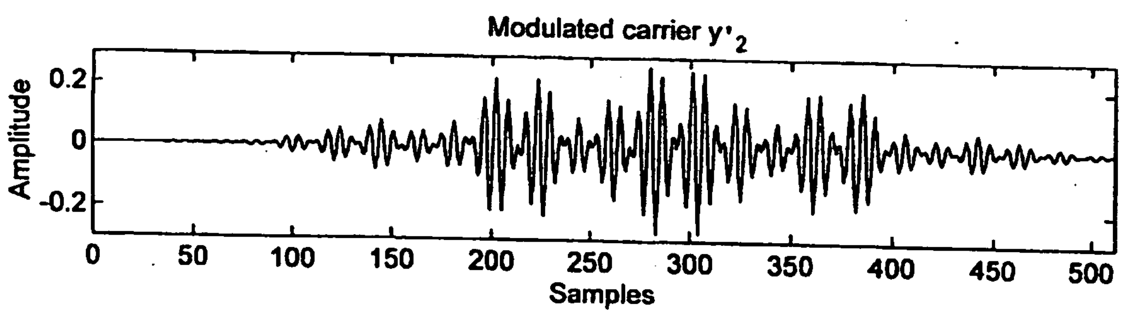


FIG. 10C

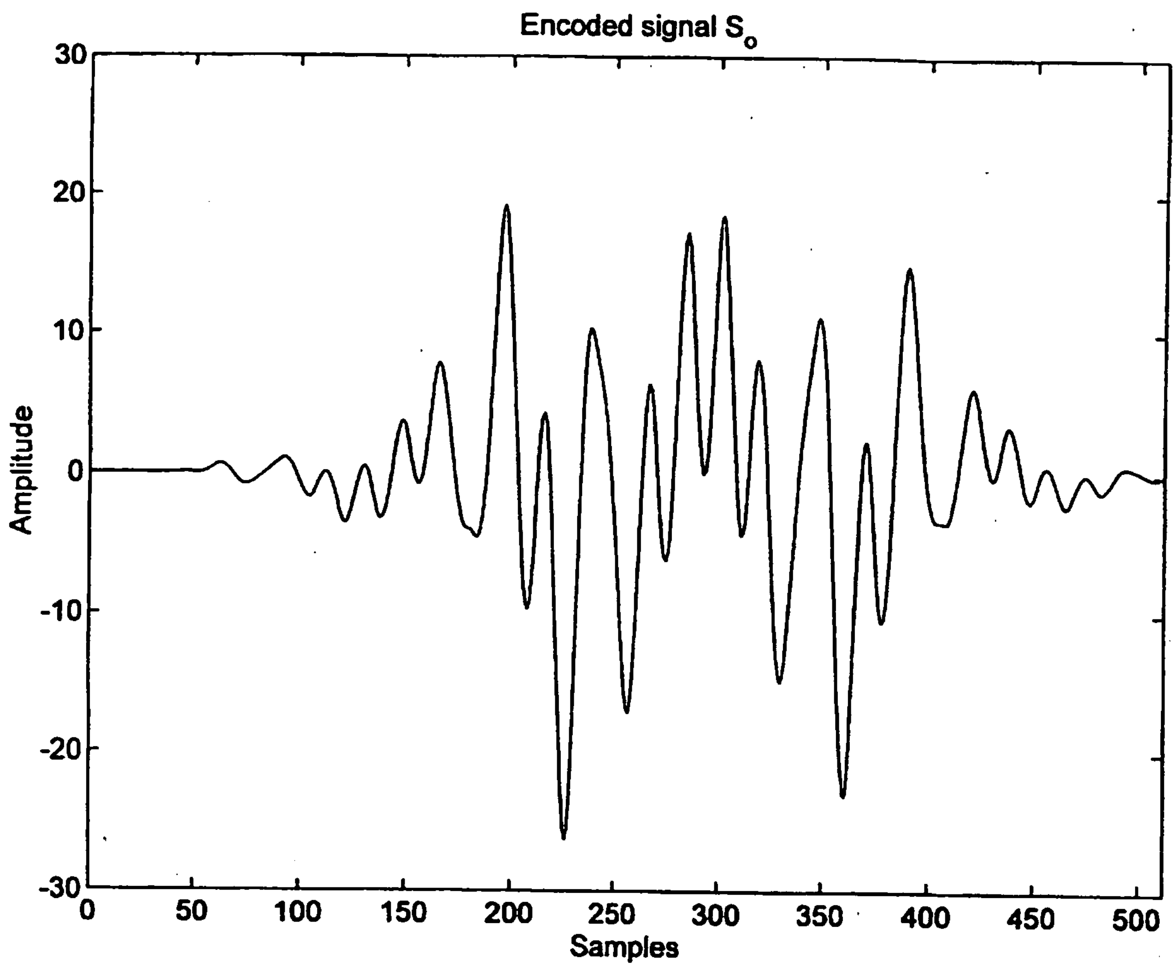

Encoded signal S_o
30
20
10
0
-10
-20
-30
Amplitude
0
50
100
150
200
250
300
350
400
450
500
Samples

FIG. 11

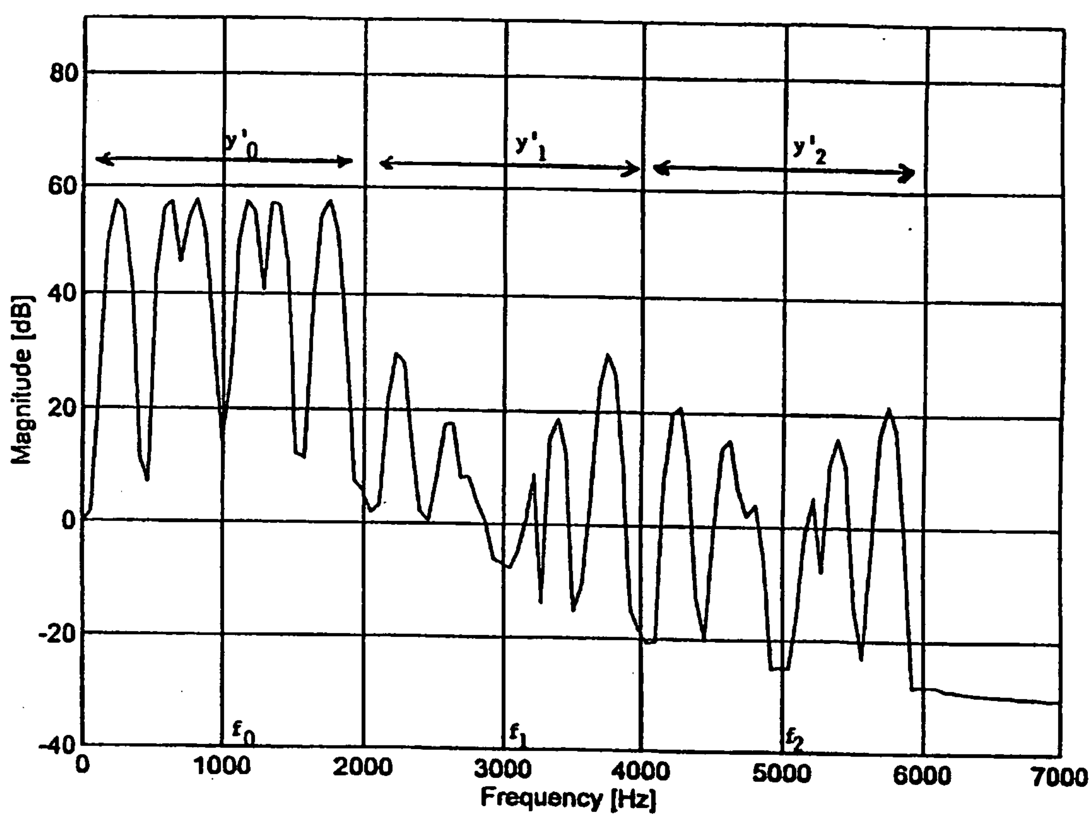

y'0
y'1
y'2
80
60
40
20
0
-20
-40
Magnitude [dB]
f0
f1
f2
0
1000
2000
3000
4000
5000
6000
7000
Frequency [Hz]

FIG. 12

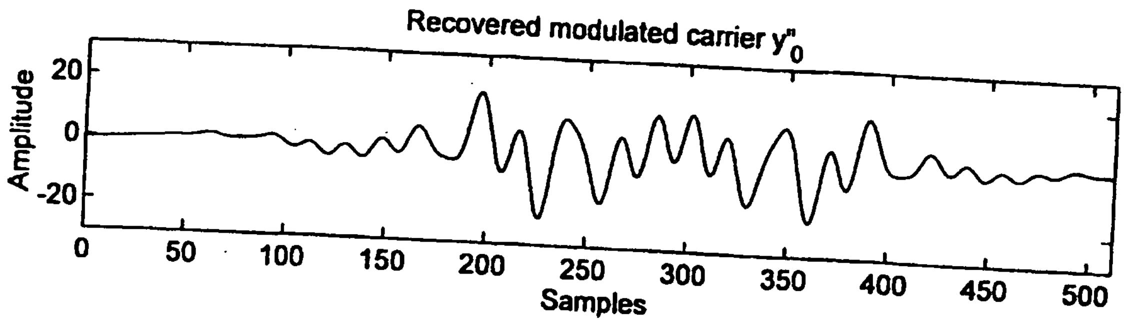


FIG. 13A

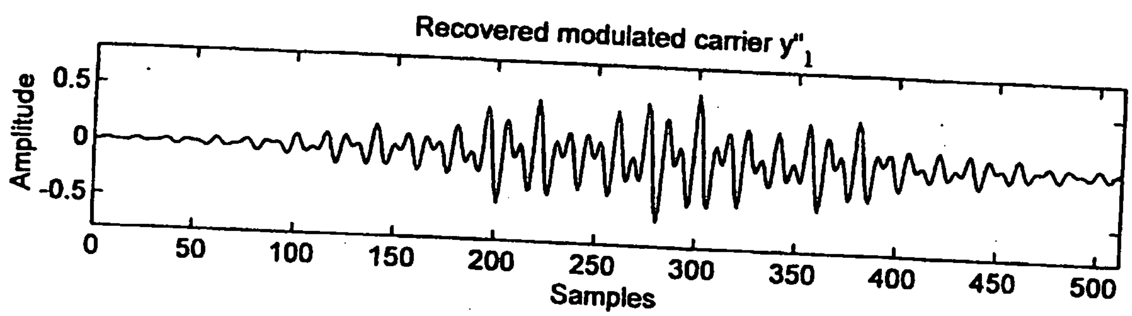


FIG. 13B

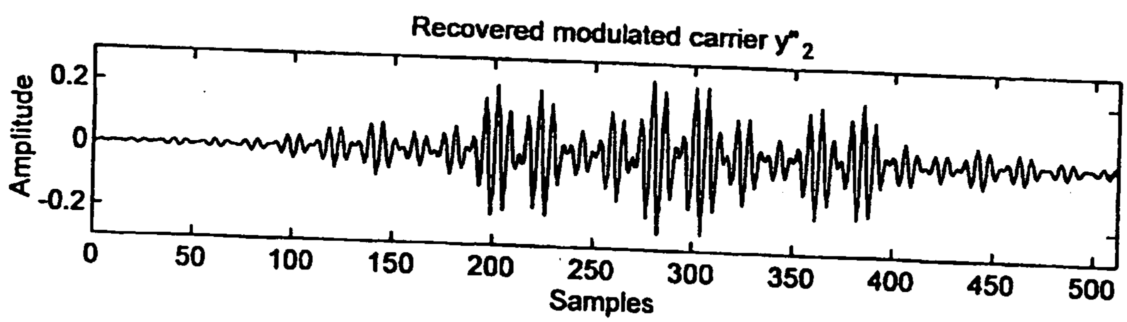


FIG. 13C

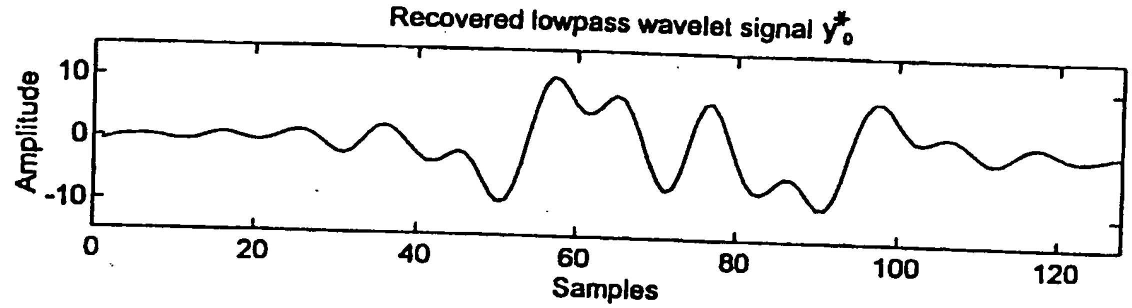


FIG. 14A

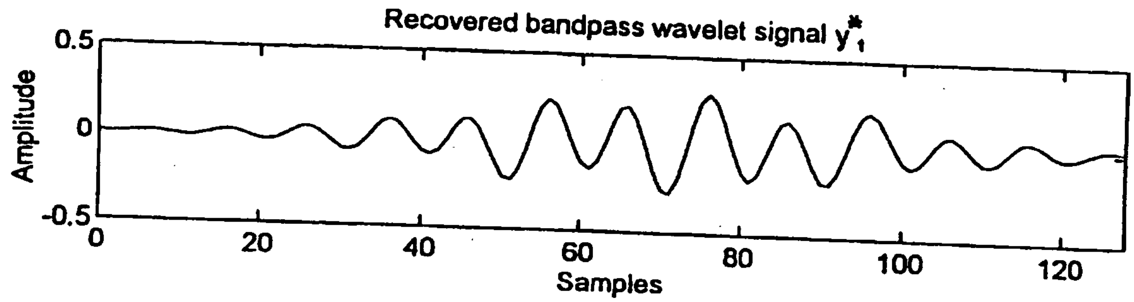


FIG. 14B

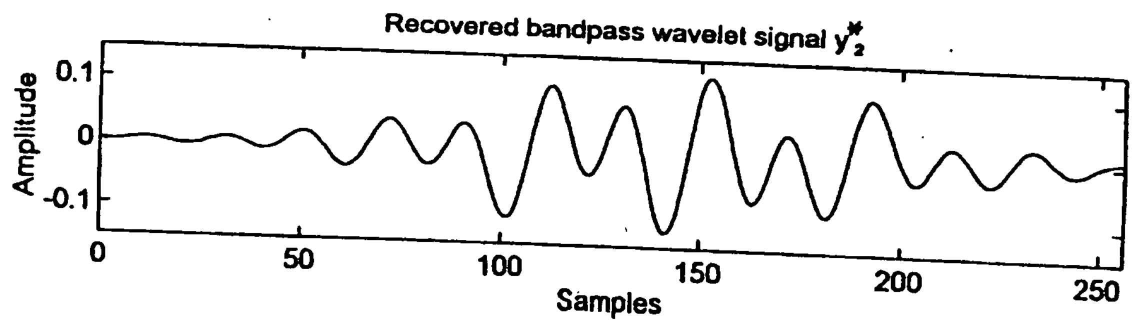


FIG. 14C

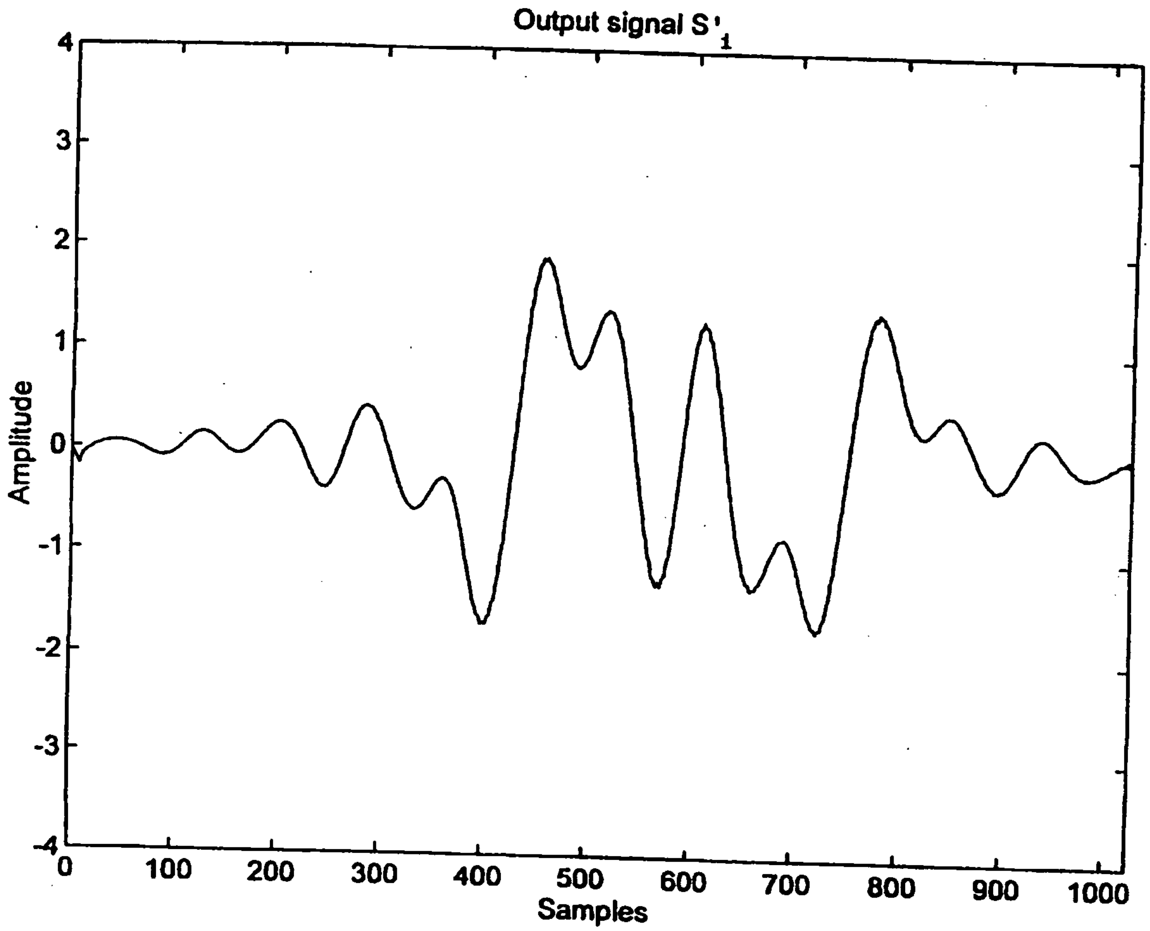

Output signal S'1
Amplitude
Samples
4
3
2
1
0
-1
-2
-3
-4
0
100
200
300
400
500
600
700
800
900
1000

FIG. 15

INTERNATIONAL SEARCH REPORT

Interr nal Application No
PCT/CA 97/00608

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04B1/66 H04L5/06

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04B H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 3 875 341 A (GASSMANN) 1 April 1975 see column 1, line 4 - line 5 see column 1, line 13 see column 1, line 27 - line 28 see column 1, line 58 - line 65 see column 2, line 20 - line 24 see column 4, line 41 - line 49 see figure 1	1,2,9, 10,17, 20,21, 28,29,35
Y		3,4,8, 11,12, 16,19, 22,23, 27-29,37
	-/--	

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
17 December 1997	29/12/1997

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Scriven, P

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	YOW-JONG LIU, YA-QIN ZANG: "Wavelet-coded image transmissionover land mobile radio channels" IEEE GLOBAL TELECOMMUNICATIONS CONFERENCE, 6 - 9 December 1992, NEW YORK, US, pages 235-239, XP000357791 see page 236, right-hand column, paragraph 2	3,4,8, 11,12, 16,19, 22,23, 27-29,37
Y	HEEGARD, SHAMOON: "High-fidelity audio compression: fractional-band wavelets" 1992 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, 23 - 26 March 1992, NEW YORK, US, pages 201-204, XP000356972 see page 201, left-hand column, paragraph 3 see page 201, right-hand column, paragraph 2	3,4,11, 12,22, 23,28,29
A	HERLEY, VETTERLI: "Orthogonal time-varying filter banks and wavelets" PROCEEDINGS OF THE IEEE INTERNATIONAL SYMPOSIUM ON CIRCUITS AND SYSTEMS, 3 - 6 May 1993, NEW YORK, US, pages 391-394, XP000410017 see figures 1,2	1,9,17, 20,28,35

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INTERNATIONAL SEARCH REPORT

.ormation on patent family members

Interr nal Application No
PCT/CA 97/00608

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